User Manual
Series
IP Phone
Version 1.2

Contents

About This Manual

This Manual provides basic information on how to install and connect 3130IF IP Phone to the network. It also includes features and functions of 3130IF IP phone components, and how to use them.

Before Getting Started

Before you can connect 3130IF to the network and use it, you must have a high-speed Internet connection installed. A high-speed connection includes such environments as DSL, cable modem, and a leased line.

1 Summary

3130IF IP Phone

3130IF IP phone is a stand-alone device, which requires no PC to make Internet calls. 3130IF IP phone supports both data and Rec/Finish thru IP network, and also provides features of conventional phone. 3130IF IP phone guarantees clear and reliable Rec/Finish quality on IP network. 3130IF IP phone can be used thru Internet phone service to make basic Internet calls. 3130IF IP is fully compatible with SIP and H.323 industry standard and can interoperate with many other SIP or H.323 compliant devices and software on the market.

2 Introduction

The 3130IF VoIP phone with the latest VoIP technology for the caller with high demands, whether at home or in the office. With a Three-line LCD display, alphanumerical caller ID and user interface with a multitude of features. High quality audio, Advanced functionality and usability and speakerphone.

Stylish and functional in design, the 3130IF IP can be used in residential, SOHO, enterprise and small to medium business service offerings including IP PBX, hosted IP telephony and IP Centrex. The 3130IF IP phone market leading technology and manufacturing proficiency to deliver an upgradeable, high quality IP telephone unparalleled in value and support.

Components Check List

1) One 3130IF IP phone
2) One Straight Ethernet cable
3) One universal power adapter, one CD

Specifications

Item			3130IF
Adapter (Input/output/frequency)			100-240VA/7.5VDC/60Hz
Port	WAN		10/100Base T RJ-45 for LAN
	LAN		10/100Base T RJ-45 for PC
Power Consumption			1.8W/1.4W
Operating Temperature			0~50℃
Relative Humidity			5~65%

Rec/Finish Features:

- Codec: G.711A/u, G.7231 high/low, G.729
- G.168 echo cancel
- Rec/Finish Gain Setting
- Jitter Buffer
- Auto latency recover
- VAD
- CNG
- RFC2833 DTFM relay
- Support H323 and SIP synchronously

H323 Features:

- H323v4
- DNS name of GK
- H323 Call forward
- H323 Fast start
- Early H245
- H245 tunneling

- H245 multiplex
- H245 DTMF/Q931 DTMF facility
- Q931 signaling port setting
- Dual GK support
- NAT transverse, CITRON
- NAT transverse, AVS
- Fast start with early media channel Rec/Finish
- Peer to Peer H323 call

SIP Features:

- RFC3261, RFC3262, RFC3666, RFC2543
- Proxy and Register
- SIP domain
- Server authentication: none, basic, MD5
- DNS name of SIP server
- SIP signaling port setting
- NAT transverse, STUN
- NAT transverse, SIP Express router
- Pubic Server/ Private server. Can connect to ISP and Private SIP server at same time
- Dual public server
- SIP INFO for DTMF, interoperate with CISCO SIP device
- Each password for each number
- SIP Call forward/transfer/holding/waiting
- Peer to peer SIP call

Networks Features:

- WAN/LAN port with Router or Bridge Mode

- Basic NAT and NAPT
- NAT ALG
- Under Bridge mode, Access internet by using NAT through PPPoE
- PPPoE for xDSL, automatically keep alive
- DHCP Client on WAN
- DHCP server on LAN
- DNS client with 2 servers IP
- DNS relay on LAN
- Auto configuration on LAN for IP and DHCP server
- SNTP
- 802.1P QOS
- 802.1q
- Firewall
- Network utilities: ping, trace route, telnet client

Call Control Features:

- Flexible Dial Map
  - ➢ Fix length
  - ➢ End with #
  - ➢ Dial Map Table
  - ➢ Dial with time out
- Call routing table for Phone Book
- Multi phone No. for same phone
- Public No. and Private No. for phone
- HOTLINE Service, Pick up phone, dial immediately
- Black list for reject authenticated call
- Empty calling No. reject service

- Limit dialing out No. list
- No Disturb
- Caller ID display
- Call forward without condition or busy
- Dial out authentication
- Auto written message

Maintenance and Management:

- Boot Monitor
- Upgrade firmware through boot monitor
- Keyboard Config
- Telnet CLI
- HTTP WEB
- FTP, TFTP upgrade firmware
- HTTP upgrade firmware
- FTP, FTP upload/download configuration file
- Security firmware upgrade, firmware digest check

AAA and Logs:

- Log level
- Telnet logs and CDR
- SysLog Logs and CDR
- FTP/TFTP CDR upload

3 Installation

3130IF IP phones are designed to look and feel like standard telephones. The following photo illustrates the appearance of a 3130IF IP phone and the use of its key buttons.

Installation

Remove the LAN cable for Internet connection from your PC and connect it to 'WAN' port of 3130IF Connect the power adapter in the box to 'Power' Find LAN cable in the box and connect between 'Lan' port and your PC (PC is not required for set up or making a call.)

4 Product Overview
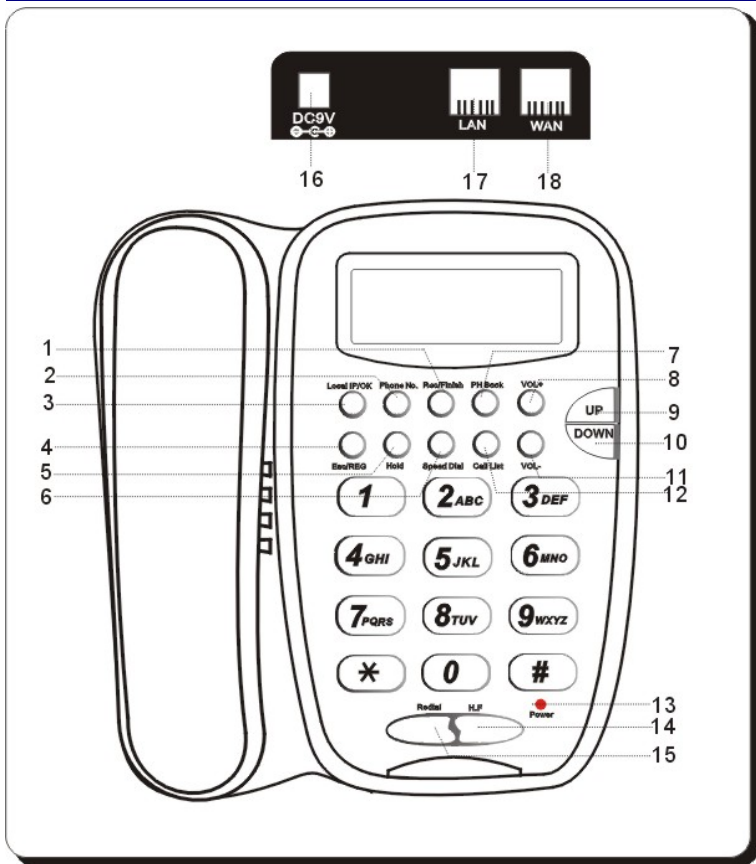
3130IF IP Phone is a next generation IP network telephone based on industry open standard SIP (Session Initiation Protocol) and H.323. Built on innovative technology, 3130IF IP Phone features market leading superb sound quality and rich functionalities at mass affordable price.

5 Basic Operations

5.1 Get Familiar with Keypad

3130IF phone has a 26-button keypad.

Key Button	Key Button Definitions
0 - 9, *, #	Digit, star and pound keys are usually used to make phone calls
Local IP/OK	Display local IP address on LCD and enter key
Phone No	Browse phone number for this unit, and can be used to modify during editing
Vol +	Increase handset/speakerphone volume
Vol -	Reduce handset/speakerphone volume
Up	Previous menu item when phone is in IDLE mode
Down	Next menu item when phone is in IDLE mode
ESC/REG	Exit and register
Hold	Temporarily hold the active call
Speed Dial	Dial speed dial number
Call List	Browse call memory
Redial	Dial a new number or Redial the number last dialed. After entering the phone number, pressing this key would force a call to go out immediately before timeout
H.F	Enter hands-free mode
Rec/Finish	Enter Rec/Finish record menu
PH Book	User can make a call directly by # button if choicing the proper person in phone number book.

5.2 Make Phone Calls

Make Calls Using Regular Phone or Extension Numbers.

There are three ways to make phone calls:

1. Pick up handset or press H.F button, and then enter the phone numbers

2. Press the Redial button directly to redial the number last called. Once pressed, the last dialed number will be displayed on the LCD as the corresponding DTMF tones are played out and an outgoing call is sent.

3. Browse the OUTGOING/INCOMING history and press the # button. Once pressed, the last dialed number will be displayed on the LCD as the corresponding DTMF tones are played out and an outgoing call is sent.

4. When the unit indicates Missed calls, press Call List and Down button to enter Miss calls menu, then press Local IP/OK button to review number. Press # button to dial out this number.

5. In ideal mode, press # button and key desired number to make pre-dial. Press # button to dial out this number.

5.3 Dial out

Pick up the phone, dial the phone number, and press "#" to call out.

5.4 Answering operation:

5.4.1 User-Defined Record and playblack:

Press "Rec/Finish" key and LCD show:

```
---REC/FINISH record---
     Received
```

Press "Down" or "Up" key and LCD show"

```
---REC/FINISH record---

    User-Defined
```

Press "Local IP/OK" key and "Up" key and LCD show:

```
--- User-Defined ---
        Rec
```

Press "Local IP/OK" key and LCD show:

```
    Press OK to Rec
```

Press "Local IP/OK" key again to begin record and press "Rec/Finish" to quite recording.

And press "DOWN" or "UP" key and LCD show:

```
--- User-Defined ---
        Play
```

Press "Local IP/OK" key to playblack User-Defined recording.

5.4.2 ICM record:

The unit will playback User-Defined recording after 5 times ringer. You can begin record ICM recording end of playback User-Defined recording.

5.4.3 Playblack ICM recording:

Press "Rec/Finish" key and LCD show:

```
---REC/FINISH record---
     Received
```

Press "Local IP/OK" key and LCD show:

```
---Received---
     New
```

Press "Local IP/OK" key and LCD SHOW:

```
--- New record ---
     Record 1
```

Press "Local IP/OK" key and LCD SHOW:

```
--- Record 1---
     Play
```

Press "Local IP/OK" key and LCD SHOW:

```
Press OK To Play
```

Press "Local IP/OK" key again to playblack.

5.5 IP distribution mode selection:

Press and hold "1" button for 5s, the LCD displays "STATIC MODE";
Press and hold "2" button for 5s, the LCD display "DHCP MODE";
Press and hold "3" button for 5s, the LCD display "PPPOE MODE".

6 Configuration with WEB

The IP Phone Web Configuration Menu can be accessed by the following URI:
http://Phone-IP-Address. The default LAN IP address is "**192.168.10.1**" and
WAN IP address is "**192.168.1.179**". If the web login port of the phone is
configured as non-80 standard port, then user need to input
http://xxx.xxx.xxx.xxx, otherwise the web will show that no server has been
found),it will be shown as follows:
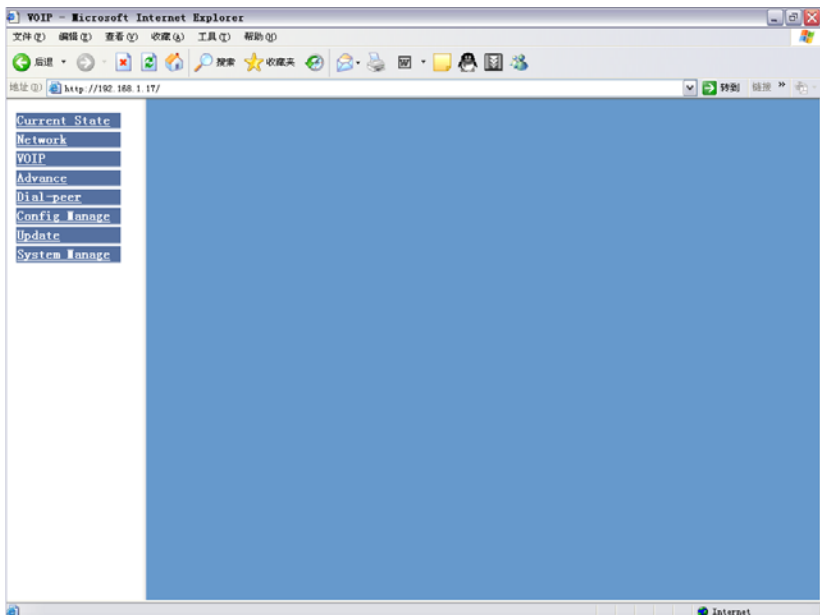
6.1 Current state

On this page user can gather information of each commonly-used parameter of
the phone, it is shown as the following figure: the network section shows the
current WAN, LAN configurations of the phone: including gaining way of WAN

IP and IP（static state, DHCP, PPPoE),MAC address, WAN IP address of the phone, LAN IP address of the phone, opening state of LAN DHCP server.

The VoIP section shows the current default signaling protocol in use, and server parameter in use of each protocol: including GateKeeper IP of H323,H323ID,whether enables register, whether has registered on GK; Register server IP of SIP, proxy server IP, whether enables register, whether has registered on register server, whether enables outbound proxy, whether enables STUN server.

The Phone Number section shows corresponding phone number of each protocol; the version number and date of issue have been shown at the end of the page.
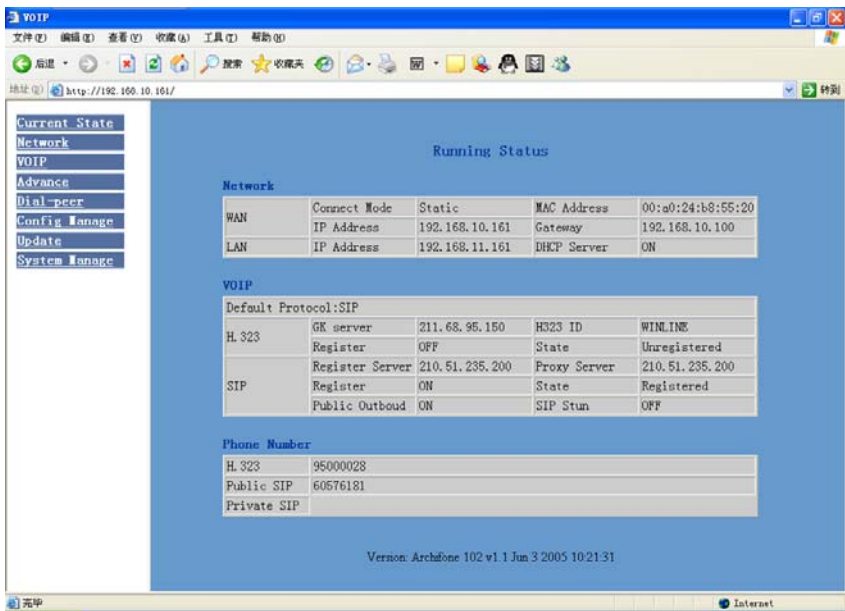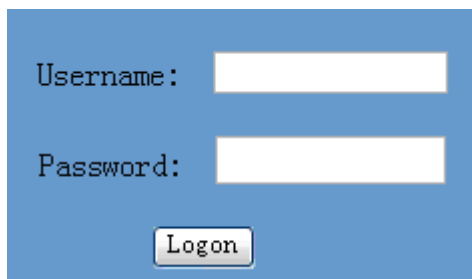
6.2 User verification

User should login before configurating dialogue machine.

Guest account: the default username and password are all" guest", user can have a browse of system.

Administrator account: the default username and password are all" admin", this user can configure the system.

Note: After inputting username and password, user press carriage return directly to enter the page.

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```
Username: [              ]

Password: [              ]

        [ Logon ]
```

6.3 Network configuration

Wide area network（**WAN**）

User can view the current network IP linking mode of the system on this page.

User will be authorized to set the network IP, Gateway and DNS if the system adopts the static linking mode.

If the system selects DHCP service in the network which is using DHCP service, IP address will be gained dynamically.

If the system selects PPPOE service in the network which is using the PPPOE service, then the IP address will be gained by the set PPPOE ISP internet and password of the account.

Note: If IP address has been modified, the web page will no longer respond owing to the modification, so new IP address should be input in the address field now.

Configuration Explanation:

Active IP	Current Netmask	MAC Address	Current Gateway
192.168.10.77	255.255.255.0	00:01:02:12:34:57	192.168.10.86

Current phone IP, subnet mask, mac address and current phone IP;

,

Select acquisition way of IP for WAN; This is single option; Configure static IP parameter for WAN;

Static	IP Address	192.168.10.77	Netmask	255.255.255.0
	Gateway	192.168.10.86	DNS Domain	voip.com
	Primary DNS	192.168.10.86	Alter DNS	192.1.1.1

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IP Address	192. 168. 10. 77

Configure static IP address;

Netmask	255. 255. 255. 0

Configure subnet mask;

Gateway	192. 168. 10. 86

Configure IP address of the phone;

DNS Domain	voip. com

Configure "DNS domain" suffix;if user input "domain" and it can't be resolved,then the phone will add and resolve the "domain" after user has input;

Primary DNS	192. 168. 10. 86

Main DNS server IP address;

Alter DNS	192. 1. 1. 1

The second DNS server IP address;

Configure PPPoE:

PPPOE Server	ANY	User	user123	Password	••••••••

Server	ANY

Service name, if PPPoE ISP has no special requirement for this name, generally is the default;

User | user123

PPPoE account;

Password | ••••••••

PPPoE password;

Configure the parameter and then click "apply" to go into effect.

Local area network (LAN)

User can make local area network (LAN) configuration on this page, when bridging mode is selected, the local area network (LAN) configuration will no longer go into effect.

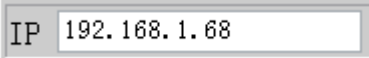## LAN Configuration

☐ Bridge Mode

| IP | 192.168.10.11 | Netmask | 255.255.255.0 |
| ☑ DHCP Service | | ☑ NAT | |

Configuration Explanation:

☐ Bridge Mode

Use bridge mode（transparent mode）:bridge mode will make the phone no

longer set IP address for LAN physical port, LAN and WAN will join in the same network;

IP 192.168.1.68

Configure LAN static IP;

Netmask 255.255.255.0

Configure LAN subnet mask;

☑ DHCP Service

Enable LAN port DHCP server; after user modify LAN IP, the phone will automatically modify the adjustment and save the configuration according to IP and subnet mask team DHCP Lease Table, user need to restart the phone to make DHCP server configuration go into effect;

☑ NAT

Enable NAT.

6.4 VOIP configuration

H.323 configuration

User can configure specific parameter of H323 signaling protocol on this page;

H323[Registered]Configuration

Default GK Addr	202.105.135.95	Alter GK Addr	211.68.95.130
Default GK Port	1719	Alter GK Port	1719
Default GK ID		Alter GK ID	
H323 ID	.ipgw.89001140	Q931 Signal Port	1720
Phone Number	89001140	GK Detect Interval	60 s
RAS Port	0	DTMF Mode	DTMF_RELAY
☑ Permit Call if not registered		☑ EARLY TALK	
☐ EARLY H245		☑ Fast Start	
☑ Enable Register		☐ Auto Detect GK	
☑ H245 Tunnel		☐ Select Multiplexing	
☐ H323 Force G7231		☑ GK Auto Swap	
☐ H323(Default Protocol)			

Apply

Configuration Explanation:

H323[Unregistered]Configuration

show H323 register state; if register successfully, there will show Registered in the square bracket, otherwise show Unregistered;

Default GK Addr   211.68.95.150

Configure default GateKeeper IP address;

Default GK Port   1719

Configure default GK port;

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Default GK ID	

Configure default GK ID; if no special requirement of GK, user don't need to fill in anything;

Q931 Signal Port	1720

The system initiates Q931 signal port,the default is 1720;

RAS Port	0

Configure the net gate RAS register port for the system; terminal user can logon to gatekeeper through RAS passage and make a request for allowing to initiate the call request. If the request has been allowed, then the gatekeeper will return a transport address（with IP address and port number）as the call signaling passage of the called party;

DTMF Mode	DTMF_RELAY
☑ Fast Start	DTMF_RELAY
	DTMF_RFC2833
☐ Auto Detect GK	DTMF_H245-STRING
	DTMF_H245-SIGNAL

Configure DTMF mode, RTP mode,RFC2833 mode,H245-string mode and H245-signal mode;

☑ Permit Call if not registered

Configure permission for no-registered call,allow to initiate call without net gate register;

☐ EARLY H245

Early245 configuration, which means that when initiating a call, the 225 message transmission begins at the same time with 245 message transmission, the default is Disable;

☐ Enable Register

Configure enable/cancel register;

☑ H245 Tunnel Configuration for transferring 245 message package to

225 message package;

☐ H323 Force G7231

Configure H323 to run the talking only by G.7231 encode, the default is Disable;

☐ H323(Default Protocol)

Configure the phone use H323 protocol as default call protocol;

☑ Fast Start

Configure quick start mode to start H323 call;

☐ Select Multiplexing

Configure multipexing of logical channel, the default is Disable;

☑ EARLY TALK

Configure the phone can receive IVR, such as the Rec/Finish prompt, dialing of

PSTN color ring;

Configure GK backup and enable GK detecting and auto-swap functions ,the phone will automatically swap to GK backup server when there is no response from default GK, and test the default GK; if the default GK recovers response, the phone will automatically swap to the default GK.

Alter GK Addr	211.68.95.130

Configure GK backup server IP;

Alter GK Port	1719

Configure server port for GK backup;

Alter GK ID	

Configure ID for GK backup;

GK Detect Interval	60	s

GK detection interval time configuration, the unit is second;

☑ GK Auto Swap

Enable the phone's auto-swap to GK;

☐ Auto Detect GK

Configure the phone to detect GK automatically.

SIP configuration

User can configure specific parameter of H323 signaling protocol on this page.

SIP[Registered] Configuration

Register Server Addr	210.51.235.200	Proxy Server Addr	210.51.235.200
Register Server Port	5060	Proxy Server Port	5060
Register Username	60576181	Proxy Username	60576181
Register Password	••••••••	Proxy Password	••••••••
Phone Number	60576181	Local SIP Port	5060
Detect Interval Time	60 seconds	Register Expire Time	33 seconds
DTMF Mode	DTMF_RFC2833	RFC Protocol Edition	RFC3261
☑ Enable Register		☐ Auto Detct Server	
☑ Enable Pub Outbound Proxy		☐ Server Auto Swap	
☑ SIP(Default Protocol)			

Configuration Explanation:

SIP[Unregistered] Configuration

show SIP register state; if register successfully, there will show Registered in the square bracket, otherwise show Unregistered;

Register Server Addr | 221.11.11.100

Configure SIP register server IP address;

Register Server Port | 5060

Configure SIP register server signal port;

Register Username	92975421

Configure SIP register account（usually it is the same with the port number that configured, some special SIP servers will have different port configurations, then the port configuration needs to be configured to be numbers, here the configuration account can be arbitrary character string）;

Register Password	●●●●●●

Configure password of SIP register account;

Proxy Server Addr	222.41.97.135

Configure proxy server IP address（usually SIP will provide user with service of proxy server and register server which have the same configuration, so the configuration of proxy server is usually the same with that of register server, but if the configurations of them are different(such as different IP addresses), then each server's configuration should be modified separately）;

Proxy Server Port	5060

Configure SIP proxy server signal port;

Proxy Username	92975421

Configure proxy server account;

Proxy Password	●●●●●●

Configure proxy server password;

Local SIP Port	5060

Configure local signal port, the default is 5060（this port will go into effect immediately, the SIP call will use the modified port for communication after modification）;

Register Expire Time	300	seconds

Configure expire time of SIP server register, the default is 600 seconds. If the expire time that server requires is more or less than that configured by the phone, the phone can automatically modify it to the recommended time limit and register;

Detect Interval Time	60	seconds

Configure detection interval time of the server, if the phone enables SIP detection server function, the phone will detect once for whether the server has response every other detection interval time;

☐ Enable Register

Configure enable/disable register;

☑ Enable Pub Outbound Proxy

Configure to enable public outbound proxy. If proxy server has been enabled, the phone will consider the user as using outbound proxy automatically. If the configuration has been disabled, the phone can still be registered to the server ,but can't make SIP call; configuration of registered call by the phone will not have impacts on SIP point-to-point call;

☑ SIP(Default Protocol)

Configure SIP of the phone as default protocol;

RFC Protocol Edition   RFC3261 ✓

Enable the phone to use protocol edition. When the phone need to communicate with phones which is using SIP1.0 such as CISCO5300 and so on, then it should be configured into RFC2543 to communicate normally. the default is to enable RFC3261;

DTMF Mode	DTMF_SIP_INFO ✓
☐ Enable Register	DTMF_RELAY DTMF_RFC2833
☑ Enable Pub Outbound	DTMF_SIP_INFO

DTMF sending mode configuration; three kinds: the above are basic configurations of SIP.

Note: If you want to register and call through server, you must configure corresponding numbers（which are usually SIP accounts）to local port, Otherwise the phone will reject for sending out register message when it considers that there is no number.

☐ Auto Detct Server

Configure automatic detection server of the phone;

☐ Server Auto Swap

Configure main and backup auto-swap server; if the phone enables main and backup server function, the automatic detection and auto-swap functions

should both be chosen;

After the aforesaid network and VoIP configurations have been configured on the phone and internetwork communication has been implemented, the user can make VoIP calls by the calling register and proxy.

SOME ISP INTERNET MAY INHIBIT THE PHONE TO REGISTER AND CANCEL THE REGISTER IN SUCCESSION, SO USER HAD BETTER NOT APPLY OR REGISTER AND CANCEL SOON IN SUCCESSION AND SUBMIT REGISTRATION REPEATEDLY. SERVER MAY STOP RESPONSE OF DIALOGUE MACHINE, THEN THE PHONE RECEIVES NO CERTIFICATION OF REGISTER/CANCEL LOGIN REQUEST AND REGISTRATION STATE WILL SHOW AS INCORRECT!

6.5 Advance configuration

Net Service configuration

User can set up Telnet, HTTP, RTP port on this page and view DHCP table.

Net Service			
HTTP Port	80	Telnet Port	23
RTP Initial Port	10000	RTP Port Quantity	200

Configuration Explanation:

HTTP Port	80

Configure web browse port, the default is 80 port, if you want to enhance system safety, you'd better change it into non-80 standard port;

| Telnet Port | 23 |

Configure telnet port, the default is 23 port;

| RTP Initial Port | 10000 |

Enable RTP initial port configuration. It is dynamic allocation;

| RTP Port Quantity | 200 |

Configure the maximum quantity of RTP port. The default is 200;

| Leased IP Address | Client hardware Address |

Leased IP-MAC correspondence table of DHCP;

※The configuration on this page needs to be saved after modified and will go into effect after restarting.

※If the Telnet, HTTP port will be modified, the port is better to be set as greater than 1024, because the 1024 port system will save ports.

※Set the HTTP port as 0,then the http service will be disabled.

SIP advanced configuration

Set SIP STUN, private and backup server, user password and so on.

SIP STUN is a kind of server that used to realize the SIP's enablement of NAT, when the STUN server IP of the phone has been configured（generally the default is 3478）and Enable SIP Stun has been selected, conventional SIP server can be used to realize the phone's penetration of NAT.

Public backup server can implement the proxy of the dialogue machine through auto-swap function when no response to public server. When the phone detect response of public server, it will auto-swap to public server. Public backup

server is redundancy backup of public server, it should have the same account with public server.

The phone's supports to two different kinds of SIP server concurrently can be implemented on private server. In this way user can register and use two different kinds of services concurrently.
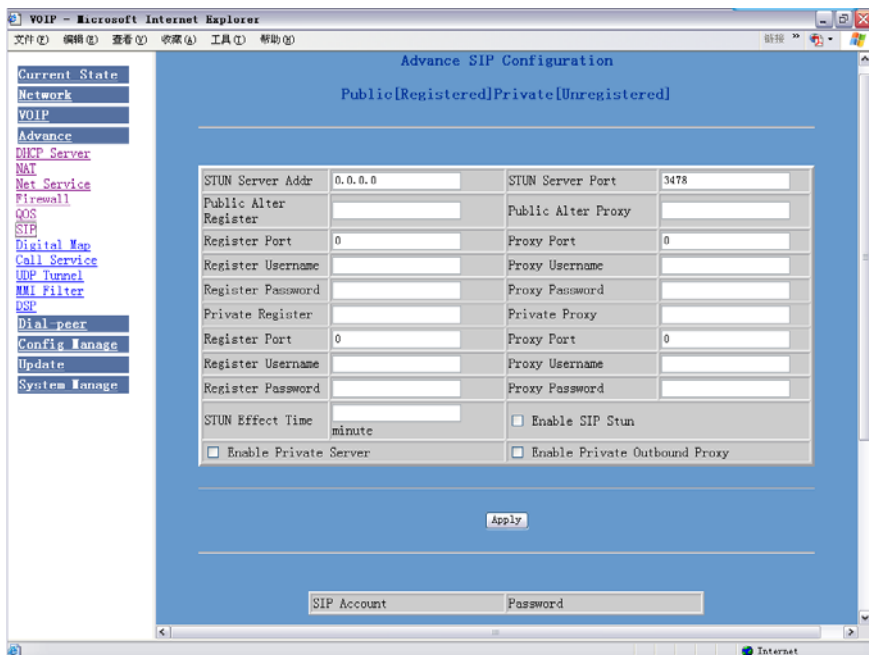
Configure explanation of private server;

show the phone whether has been registered on public server or private server;

STUN Server Addr | 0.0.0.0

Configure IP address of SIP STUN server;

STUN Server Port | 3478

Configure port of SIP STUN.

STUN can support SIP terminal's penetration to NAT in the inner-net. In this way, as long as there is conventional SIP proxy and a STUN server placed in the public net, it will do; but STUN only supports three NAT modes: FULL CONE, restricted, port restricted;

Public Alter Register	10.1.1.11	Public Alter Proxy	0.0.0.0
Register Port	5060	Proxy Port	5060
Register Username	1234	Proxy Username	1234
Register Password	••••	Proxy Password	••••

Public backup server configuration; the specific configuration parameter has the same meaning with public server. It should be noted that the username and password should be the same with the public main server;

Private Register	210.25.132.124	Private Proxy	210.25.132.124
Register Port	5060	Proxy Port	5060
Register Username		Proxy Username	
Register Password		Proxy Password	

Private server configuration. specific configuration parameter has the same meaning with public server;

STUN Effect Time |
minute

Interval time for STUN's detection on NAT type, the unit is minute;

☐ Enable SIP Stun

Configure enable/disable SIP STUN;

☐ Enable Private Server Register

Configure permit/deny private server register;

☐ Enable Private Outbound Proxy

Configure enable/disable private outbound proxy;

If user has accounts of a certain SIP server and each account has different password, then user should add each account and its corresponding password to the account& password table.

SIP Account	Password
1000	1000

Configure display of account & password list;

Click Add to add account and password, it is shown as the following figure:

SIP Account	
SIP Password	
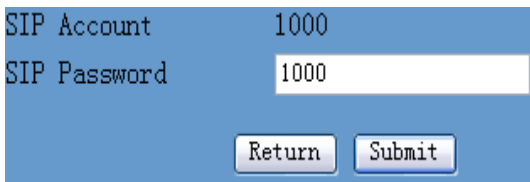
Return   Submit

Click submit to submit the configuration, click return to cancel the configuration and return;

Delete   1000 ∨

Select accounts that you want to delete from the drop-down menu, click delete.

Modify   1000 ∨   Load

Select drop-down menu to select accounts that want to modify, click load to load the configuration and then click modify;

SIP Account	1000
SIP Password	1000
	Return Submit

Accounts to be Modified, read-only;
Passwords to be modified;
Click submit to submit, click return to cancel the modification and then return.

Value added service configuration

On this page, user can set value added services such as hot-line, call forwarding, call transfer (CT),call-waiting service three way call, blacklist, out-limit list and so on.

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Call Service

Hotline	

Call Forward	◉ Off ◯ Busy ◯ No Answer ◯ Always

| | Faraway Protocol:H323 Number | | Addr | | Port | 1720 |

| | Faraway Protocol: SIP Number | | Addr | | Port | 5060 |

☐ No Disturb	☐ Ban Outgoing
☐ Enable Call Transfer	☐ Enable Call Waiting
☐ Enable Three Way Call	☐ Accept Any Call
☐ Auto Answer	☐ Enable Voice Record
☑ User-Defined Voice	☐ Incoming Record Playing
20 No Answer Time(seconds)	

Configuration Explanation:

Hotline	

Configure hot-line number of the port. With this number of the port, this hot-line number will be dialed automatically as soon as off-hook and user can's dial any other number;

Call Forward	◉ Off ◯ Busy ◯ Always

Call forwarding. The default is Disable; when busy is selected, if the number dialed is engaged after the phone has received a call, then it will automatically transfer to the configured number according to the following configuration; when always is selected, then the phone will directly transfer all the numbers that dial to this port to the configured numbers;

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| Faraway Protocol:H323 | Number | | IP | 0.0.0.0 | | Port | 1720 |
| Faraway Protocol:SIP | Number | | IP | 0.0.0.0 | | Port | 5060 |

number IP configuration of call transfer (CT);

☐ Enable Call Waiting

Configure enable/disable call waiting service; after it is enabled, user can hold calls of the other party by hooking, with hooking again, the hold call can go on;

☐ Enable Call Transfer

Configure enable/disable call transfer (CT); after it is enabled, user accept calls, with hooking and dial directly, the phone will transfer the calls according to the above   configurations of the port number IP images;

☐ Enable Three Way Call

Configure enable/disable three way call; user can call the other part as the call origination, after talking, make hooking to hold this part and then press * key to hear the dialing tone, after call completion to the third party, hooking again to recover the talk with the second part, then the three way call concurrently;

☐ Enable Voice Record

Configure enable/disable Enable Rec/Finish Record, then no body answer the call, the phone will into the answering function;

☑ User-Defined Voice

Configure enable/disable User-Defined Rec/Finish, then enable Rec/Finish record, the phone will auto request user leave message.

After the aforesaid configuration has been done, click apply to make them go into effect.

Black List			
	Add	⌄	Delete

Configure add/delete blacklist. If user doesn't want to answer a certain number, please add this number to the list, and then this number will be unable to get through the phone.

Limit List			
	Add	⌄	Delete

Configure out-limit list; for example, if user don't want the phone to dial a certain number, please add the number to this table, and the user will be unable to get through this number.

DSP configuration

On this page, user can set speech coding, IO volume control, cue tone standard, caller ID standard and so on.

DSP Configuration				
Coding Rule	g723-r63 ⌄	Handdown Time	200	ms
Input Volume	5 (1-5)	Output Volume	5	(1-9)
Handfree Volume	5 (1-9)			

Apply

Configuration Explanation:

Output Volume	5	(1-9)

Configure output volume;

Input Volume	5	(1-5)

Configure input volume;

Handfree Volume	5	(1-9)

Configure handsfree volume;

Handdown Time	400	ms

Configure handdown time, that is, if the hooking time is shorter than this time, then the gateway will not consider the user has handdown.

6.6 Number binding configuration
Number IP table configuration

Function of number IP table is one way to implement the phone's calling online, and the calling of the phone will be more flexible by configurating the number IP table.   For example, user know the other party's number and IP and want to make direct call to the party by point-to-point mode: the other party's number is 1234,make a configuration of 1234 directly ,then the phone will send the called number1234 to the corresponding IP address; Or set numbers with prefix matching pattern, for example, user want to make a call to a number in a certain region（010）,user can configure the corresponding number IP as 010T— protocol— IP, after that, whenever user dial numbers with 010 prefix(such as 010－62201234),the call will be made by this rule.

Bases on this configuration, we can also make the phone use different accounts and run speed calling without swap.

When making deletion or modification, select the number first and click load, then click Modify and complete the operation.

Dial-Peer						
Number	Call Mode	Destination	Port	Alias	Suffix	Del length
9T	h323	0.0.0.0	1720	del	no suffix	1
0T	sip	0.0.0.0	5060	del	no suffix	1

Add   Delete 0T ▾   Modify 9T ▾ Load

Configuration Explanation:

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Number	Call Mode	Destination	Port	Alias	Suffix	Del length
0T	lifeline	0.0.0.0	0	no alias	no suffix	0
9T	sip	0.0.0.0	0	no alias	no suffix	0
1T	h323	0.0.0.0	1720	no alias	no suffix	0
8T	sip	255.255.255.255	5060	del	no suffix	1

Display of calling number IP image list;

Add
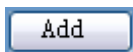
Click Add, the following figure will be shown at the lower part of the page, of which;

Phone Number 010T

It is to add outgoing call number, there are two kinds of outgoing call number setup: One is exactitude matching, after this configuration has been done, when the number is totally the same with the user's calling number, the phone will make the call with this number's IP address image or configuration; Another is prefix matching（ be equivalent to PSTN's district number prefix function）,if the previous N bits of this number are the same with that of the user's calling number(the prefix number length),then the phone will use this number's IP address image or configuration to make the call. When configurating the prefix matching, letter "T" should be added behind the prefix number to be distinguished from the exactitude matching; the longest length is 30 bits.

Call Mode sip

Configure the calling mode:H323 and SIP;

Destination 192.168.10.11

Configure destination address, if it is point-to-point call, then input the opposite terminal's IP address, it can also be set as domain name and resolved the specific IP address by DNS server of the phone. If no configuration has been made, then the IP will be considered as 0.0.0.0. This is an optional configuration item;

Port(optional)

Configure the other party's protocol signal port, this is optional configuration item :when nothing is input, then the default of h323 protocol is 1720,the default of sip protocol is 5060;lifeline required no configuration of this item, shown as 0;

Alias(optional)

Configure alias, this is optional configuration item: it is the number to be used when the other party's number has prefix; when no configuration has been made, shown as no alias;

Suffix(optional)

Configure suffix ,this is optional configuration item: it is the additive dial-out number behind the number; when no configuration has been made, shown as no suffix;

Delete Length
(optional)

Configure the replacing length, replace the number that user input according to this length; this is optional configuration item.

Of which the alias can be divided into four types, it should be combined with replacing length to make the setup:

Add: xxx, add xxx before number. in this way it can help user save the dialing length;

All: xxx, the number is all replaced by xxx; speed dialing can be implemented, for example, user configure the dialing number as 1, with the configuration "all" , the actual calling number will be replaced;

Del: delete n bit in the front part of the number, n can be decided by the replacing length; this configuration can decide the protocol for appointed number;

Rep: xxx, n bit in the front part of the number will be replaced. n is decided by the replacing length. For example, user want to dial PSTN（010－62281493） by VoIP's Rec/Finish over service, while actually the called number should be 8610 － 62281493,then we can configure called number as 010T,then rep:8610,and then set the replacing length as 3. So that when user make a call with 010 prefix, the number will be replaced as 8610 plus the number and then sent out. It is a convenient thinking mode for user to make a call;
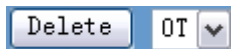
Delete | 0T ∨

Delete selective number IP image;

Modify | 0T ∨ | Load | If user want to modify a certain current number image,first select in the drop-down menu and then load the image parameter of the said number, click modify to make modification; of which:

Phone Number 9T

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this is the modified number. read-only;

Call Mode sip

To modify call mode;

IP or Domain (optional) 0.0.0.0

To modify destination address; this is optional configuration item;

Port(optional) 0

To modify destination phone port;this is optional configuration item;

Alias(optional) no alias

To modify alias; this is optional configuration item;

Suffix(optional) no suffix

To modify suffix; this is optional configuration item;

Delete Length (optional) 0

To modify replacing length（if rep and del of alias have been configured）;

Return   Submit

Click submit to go into effect; click return to cancel configuration and return.

The basic application of the number IP table has been introduced, now let me introduce how to configure IP table of number to implement configuration of using multi-accounts concurrently:

For example, now user has a H323 account and two SIP accounts, then under

the default condition, user can only make calls by the default protocol. Configure the number IP table to select the call protocol, then user don't need to select default protocol before making calls every time.

The configuration process will not be repeated, now I will mainly introduce what kind of number IP image can implement this function.

By configuration, Image table as follows will be gained:

Number	Call Mode	Destination	Port	Alias	Suffix	Del length
9T	sip	0.0.0.0	5060	del	no suffix	1
8T	sip	255.255.255.255	5060	del	no suffix	1
7T	h323	0.0.0.0	1720	del	no suffix	1

Image of 9T means when user configure public SIP server and register, then user just need to add a"9"before the calling number whenever making a call by public SIP;

Image of 8T means when user configure private server and register, then user just need to add a"8"before the calling number whenever making a call by private SIP;

Image of 7T means when user configure h323 server and register, then user just need to add a"7"before the calling number whenever making a call by H323 GK.

6.7 Save and Clear configuration

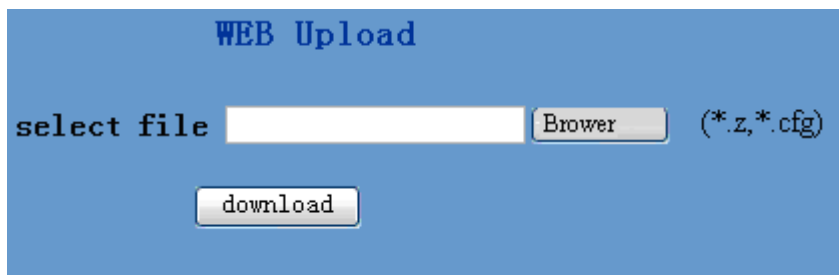User can save the current configuration on this page.

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Save Configuration

Save
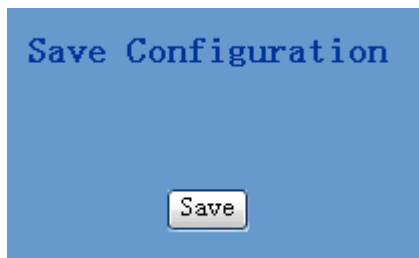
The system configuration can be set as factory default configuration on clear config page and the phone will restart automatically.

Clear Configuration

Clear

The device will reboot and use default configuration !
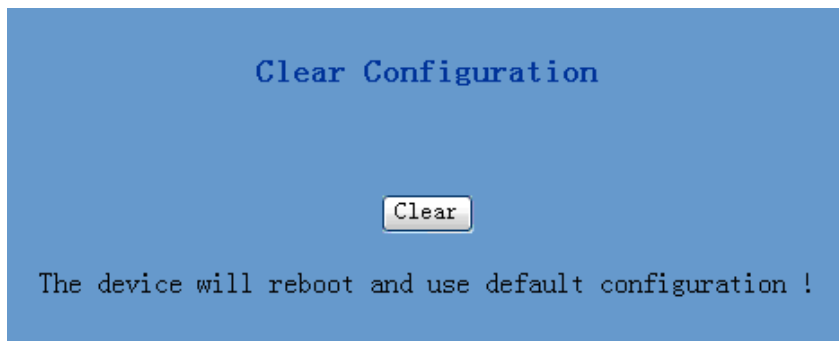
6.8 Upgrade on-line

Upload WEB page

On this page, user can select the upgrade document (firmware or config file) on hard disk of the computer directly to run the system upgrade. After the upgrade has been completed, restart the phone and it will be usable at once.

FTP download

On this page, user can upgrade system and configure files by FTP or TFTP mode.

FTP Download

Server	
Username	
Password	
File name	
Porotocol	FTP ∨

[Image Update]   [Config Upload]   [config Download]

Configuration Explanation:

Server	

Configure upload or download FTP/ TFTP server IP address;

Username	

Configure username of the upload or download FTP server. If user select TFTP mode, username and password are not required to be configured;

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Password	

Configure upload or download of FTP server password;

File name	

Configure upload or download system upgrade document or system layout file name. It should be noted that system file take .dlf as suffix, configuration files take .cfg as suffix;

Porotocol	FTP ∨
	FTP
	TFTP

Select server type;

Image Update

Click image update button, the phone will upgrade system file;

Config Upload

Click config upload button, the phone will upload its configuration files to FTP/TFTP server and save with names of user-defined configuration files;

config Download

Click config download button, the phone will download configuration files of FTP/TFTP server to the phone and the configuration will go into effect after restarting;

Configuration files WEB download

On this page, user can directly select the configuration files on the hard disk of

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the computer, and then make modification to the system configuration, after the download, restart the phone and the configuration will go into effect.

6.9 System management

Account management

On this page, user can add and delete users according to own needs and can modify user's authorities there have been.

Account Configuration	
User Name	User Level
admin	Root
guest	General

Add   Delete   guest ▾   Modify   admin ▾   Load

Configuration Explanation:

User Name	User Level
admin	Root
guest	General

display of phone user account list;

Add

To add phone account; it will be shown at lower part of page as the following figure, of which:

Add new accounts;

As account level; root possesses authorities to modify configuration, general possesses read-only authority; additive account;

As second confirmation of password, to ensure correct setup of password;

Click submit to go into effect; click return to cancel configuration and return.

Select users that you want to delete in the drop-down menu, click Delete.

To modify the chosen accounts, need to select account first, click load again and then click modify, it will be shown at lower part of page as the following figure, of which:

The modified username;

Modify user authorities;

Modify user password;
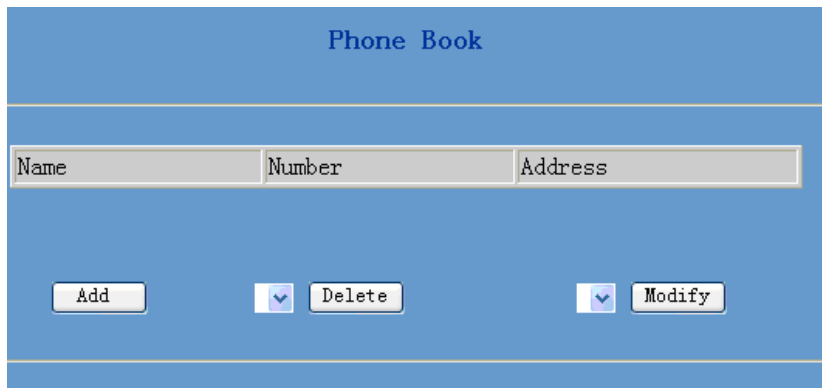Make confirmation of the modified user password;

Owing to the phone's default account: accounts of the administrator

level-admin and the ordinary level－guest are all weak account and weak password, the username and password will be easily to be guessed on public network, so the user had better modify the administrator and ordinary user.

Enter with manager level when making modification, create a administrator account and a browse account (you'd better not set the name as admin, administrator, guest, etc.),set password and then save configuration, entering with new manager account, delete default manager and browse account and save configuration, security will be enhanced!

Phone book configuration

On this page, user can save and configure telephone book.

Phone Book		
Name	Number	Address
Add	[ ] Delete	[ ] Modify

VPN configuration

On this page, user can save and configure VPN setting.

Current State
Network
VOIP
Advance
DHCP Server
NAT
Net Service
Firewall
QOS
SIP
Digital Map
Call Service
MMI Filter
DSP
VPN
Dial-peer
Config Manage
Update
System Manage

VPN Tunnel

VPN Server Addr	0.0.0.0	VPN Server Port	80
Server Group ID	VPN	Server Area Code	12345
☐ Enable VPN Tunnel		Out GK Addr	0.0.0.0

Apply

Tunnel List

Add   ▼   Delete

VPN Server Addr   0.0.0.0

Configure VPN server address;

VPN Server Port   80

Configure VPN server port;

Server Group ID   VPN

Configure VPN server group ID;

Server Area Code   12345

Configure VPN server area code;

Configure enable/disable VPN tunnel;

Configure out GK address.

Time zone configure

On this page, user can save and configure time zone setting.

Configure the desired time zone.